IN THE CLAIMS:

The text of all pending claims, (including withdrawn claims) is set forth below. Cancelled and not entered claims are indicated with claim number and status only. The claims as listed below show added text with underlining and deleted text with ~~strikethrough~~. The status of each claim is indicated with one of (original), (currently amended), (cancelled), (withdrawn), (new), (previously presented), or (not entered).

Claims are repeated below, without amendment.

1. (PREVIOUSLY PRESENTED) A wide-band speech coder comprising:

a speech characteristic classification unit, which stipulates a characteristic of speech corresponding to a current frame statistically using an open-circuit pitch value and a linear prediction coefficient in which a wide-band speech signal to be coded is perceptual weight filtered;

an adaptive codebook retrieving unit, which retrieves a pitch delay value around the open-circuit pitch value, calculates a pitch gain value, generates an adaptive codebook contribution signal corresponding to the retrieved pitch delay value, and outputs a difference between the generated adaptive codebook contribution signal and the perceptual weight filtered signal as a first fixed codebook target signal;

a first fixed codebook retrieving unit, which obtains a first fixed codebook index that can express the first fixed codebook target signal most properly, and a first fixed codebook gain value, generates a first fixed codebook contribution signal corresponding to the retrieved index, and outputs a difference between the first generated fixed codebook contribution signal and the first fixed codebook target signal as a second fixed codebook target signal;

a second fixed codebook retrieving unit, which includes at least two second fixed codebooks according to a speech characteristic, selects a second fixed codebook according to the speech characteristic, and retrieves second fixed codebook index that can express the second fixed codebook target signal most properly, and second fixed codebook gain values; and

a parameter multiplexer, which quantizes and multiplexes the speech characteristic information, the pitch delay value, the pitch gain value, the first fixed codebook index, the first fixed codebook gain value, the second fixed codebook index, and the second fixed codebook gain values, makes them as a bit stream, and transmits the bit stream to an external speech decoding terminal.

2. (PREVIOUSLY PRESENTED) The wide-band speech coder of claim 1, wherein the second fixed codebook is composed of an algebraic codebook and a random codebook, and the second fixed codebook retrieving unit retrieves the random codebook in fricative sound or affricate section and retrieves the algebraic codebook in other sections.

3. (PREVIOUSLY PRESENTED) The wide-band speech coder of claim 1, wherein the second fixed codebook is composed of an algebraic codebook and a random codebook, and the second fixed codebook retrieving unit retrieves the random codebook in an unvoiced sound section and retrieves the algebraic codebook in a voiced sound section.

4. (PREVIOUSLY PRESENTED) The wide-band speech coder of claim 1, wherein the second fixed codebook gain values include all gain values of each of the second fixed codebooks.

5. (PREVIOUSLY PRESENTED) The wide-band speech coder of claim 1, wherein the second fixed codebook gain values include a second standardized fixed codebook gain value and a ratio of the second standardized fixed codebook gain value and gain values of other second fixed codebooks.

6. (PREVIOUSLY PRESENTED) A wide-band speech coding method comprising:

stipulating a characteristic of speech corresponding to a current frame statistically using an open-circuit pitch value and a linear prediction coefficient in which a wide-band speech signal to be coded is perceptual weigh filtered;

obtaining a pitch delay value around the open-circuit pitch value and a pitch gain value and generating a difference between an adaptive codebook contribution signal corresponding to the obtained pitch delay value and the perceptual weight filtered signal as a first fixed codebook target signal;

obtaining a first fixed codebook index that can express the first fixed codebook target signal most properly, and a first fixed codebook gain value and generating a difference between a first fixed codebook contribution signal generated using the first obtained fixed codebook index and the first fixed codebook gain value and the first fixed codebook target signal as a second fixed

codebook target signal;

selecting and retrieving a second fixed codebook retrieving unit from a plurality of second fixed codebooks classified according to a speech characteristic, according to speech characteristic information and retrieving second fixed codebook index that can express the second fixed codebook target signal properly, and second fixed codebook gain values; and

quantizing and multiplexing the speech characteristic information, the pitch delay value, the pitch gain value, the first fixed codebook index, the first fixed codebook gain value, the second fixed codebook index, and the second fixed codebook gain values, making them as a bit stream, and transmitting the bit stream to an external speech decoding terminal.

7. (PREVIOUSLY PRESENTED) The wide-band speech coding method of claim 6, wherein the second fixed codebook is composed of an algebraic codebook and a random codebook, and in fricative sound or affricate section, the random codebook is retrieved, and in other sections, the algebraic codebook is retrieved, such that the second fixed codebook indices and the second fixed codebook gain values are obtained.

8. (PREVIOUSLY PRESENTED) The wide-band speech coding method of claim 6, wherein the second fixed codebook is composed of an algebraic codebook and a random codebook, and in an unvoiced sound section, the random codebook is retrieved, and in a voice sound section, the algebraic codebook is retrieved, such that the second fixed codebook indices and the second fixed codebook gain values are obtained.

9. (PREVIOUSLY PRESENTED) The wide-band speech coding method of claim 6, wherein the second fixed codebook gain values include all gain values of each of the second fixed codebooks.

10. (PREVIOUSLY PRESENTED) The wide-band speech coding method of claim 6, wherein the second fixed codebook gain values include a second standardized fixed codebook gain value and a ratio of the second standardized fixed codebook gain value and gain values of other second fixed codebooks.

11. (PREVIOUSLY PRESENTED) A wide-band speech decoder comprising:

a parameter demultiplexer, which demultiplexes a bit stream transmitted from an external wide-band speech coder, including parameter;

an adaptive code vector generator, which obtains an adaptive code vector corresponding to an adaptive codebook pitch delay value and an adaptive codebook pitch gain value;

a first fixed code vector generator, which obtains a first fixed code vector corresponding to a first fixed codebook index and a first fixed codebook gain value;

a second fixed code vector generator, which selects a second fixed codebook from a plurality of second fixed codebooks using speech characteristic information and obtains a second fixed code vector corresponding to the second fixed codebook index and the second fixed codebook gain value;

an adder, which adds the adaptive code vector and the first and second fixed code vectors to one another and generates an excitation signal, and

wherein the excitation signal is linear prediction synthesis filter processed and post-processing filter processed and is generated as a speech synthesis signal.

12. (PREVIOUSLY PRESENTED) A wide-band speech decoding method comprising:

de-multiplexing a bit stream transmitted from an external wide-band speech coder, including parameters;

retrieving an adaptive codebook and obtaining an adaptive code vector corresponding to an adaptive codebook pitch delay value and an adaptive codebook pitch gain value;

retrieving a first fixed codebook and obtaining a first fixed code vector corresponding to a first fixed codebook index and a first fixed codebook gain value;

selecting and retrieving a second fixed codebook from a plurality of second fixed codebooks using speech characteristic information and obtaining a second fixed code vector corresponding to the second fixed codebook index and the second fixed codebook gain value;

adding the adaptive code vector and the first and second fixed code vectors to one another and generating an excitation signal; and

linear prediction synthesis filter processing and post-processing filter processing the excitation signal and generating the excitation signal as a speech synthesis signal.

13. (PREVIOUSLY PRESENTED) The wide-band speech decoder of claim 11, wherein the parameters are at least any one of speech characteristic information, an adaptive codebook pitch delay value, an adaptive codebook pitch gain value, a first fixed codebook index, a first fixed codebook gain value, second fixed codebook indices, and second fixed codebook gain values.

14. (PREVIOUSLY PRESENTED) The wide-band speech decoder of claim 12, wherein the parameters are at least any one of speech characteristic information, an adaptive codebook pitch delay value, an adaptive codebook pitch gain value, a first fixed codebook index, a first fixed codebook gain value, second fixed codebook index, and second fixed codebook gain values.

15. (PREVIOUSLY PRESENTED) A wide-band speech decoding method comprising:

de-multiplexing a bit stream transmitted from an external wide-band speech coder, including parameters;

retrieving an adaptive codebook and obtaining an adaptive code vector corresponding to an adaptive codebook pitch delay value and an adaptive codebook pitch gain value;

retrieving a first fixed codebook and obtaining a first fixed code vector corresponding to a first fixed codebook index and a first fixed codebook gain value;

selecting and retrieving a second fixed codebook from a plurality of second fixed codebooks using speech characteristic information and obtaining a second fixed code vector corresponding to the second fixed codebook index and the second fixed codebook gain value;

adding the adaptive code vector and the first and second fixed code vectors to one another and generating an excitation signal; and

linear prediction synthesis filter processing and generating the excitation signal as a speech synthesis signal.